Detailed End Point IVT Test Plan and Report for ALGO 8186 SIP Horn Speaker and CUCM 12.0

Test Result	PASS
Test Date	27th June 2018
Product Name	Algo 8186 SIP Horn Speaker
Product Version # (must be generally available)	1.6.1_rc1
Unified Communications Manager Version	12.0
Product Type(Billing, Voice Recording, phone apps etc):	Endpoint
API/Protocol(s) Used	SIP
Partner IVT Contact Name:	Paul Zoehner
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IVT Lab Location (EMEA or US):	India
Partner Main Support Number	
Partner Main Support Email	

Revision History

Revision	Author	Date	Comment
1.0	tekVizion	June 21st-2018	Initial draft for Cisco review
1.1	Ashwin George	June 28th 2018	Updated the Test results

Table of Contents

1 Introduction

This document is the detailed Interoperability Verification Test Plan and Report for CUCM 12.0 and Algo 8186 SIP Horn Speaker.

1.2 Entry Criteria

Before testing can begins 3rd party partner shall run this entire test plan in their lab and verify that results. If there are any test cases not supported, not applicable or are not successful, the partner should consult with tekVizion test team. Once testing has been initiated, the device under test is considered frozen for certification testing purposes. No software/firmware load can be changed during the testing period. However, configuration can be modified to accommodate testing.

1.3 Exit Criteria

To be deemed certified as configured, the devices under test should have zero severity 1 and severity 2 defects and up to two severity 3 defects detected.

If a severity 1 or 2 failure occurs, irrespective of who is responsible for the problem (Cisco or the 3rd party product) the testing is considered unsuccessful.

Table 1 Defect Severity Level Description

Severity	Description
1	Catastrophic - Common circumstance causes the entire system or a major subsystem to stop working affects other areas/devices no workaround
2	Severe- Important functions are unusable does not affect other areas/devices no workaround
3	Moderate - Very unusual circumstances cause failure minor feature doesn't work at all there's a low impact workaround

If any tests fail, the configuration will be verified to resolve the issue. If the issue cannot be resolved, the tester will attempt to continue testing if possible. If the testing cannot proceed without this problem being resolved, the testing is considered complete and the devices under test are deemed not certified.

The following procedures are followed when testing fails:

- Preliminary analysis is made to determine the source of the problem. If the problem is related to a device under test, then the problem is reported to that partner. If the problem is deemed Cisco related, the problem will be reported to Cisco, but the partner is responsible to open a TAC case with Cisco developer services. Partner should provide the TAC case number to the test team so they can document it in the report.
- If testing can continue past this failure, the other test cases will be tested and verified for pass or fail. If the testing cannot progress past this problem, testing will be halted and a final test report submitted to Partner and Cisco.
- All problems and resolutions encountered during testing are documented in the final test report.
- If a severity 1 failure occurs, irrespective of whom is responsible for the problem (Cisco or the 3rd party product), the testing is considered unsuccessful.

Any deviations of the test execution or problem acceptance are documented in the test report.

Note: The Cisco approval process may increase/decrease the severity level of the defect after the test cycle, if considered necessary.

2 Product Overview

The 8186 is a PoE outdoor rated horn speaker that is a 3rd party SIP compliant endpoint. Compatible with Cisco hosted or premise VoIP communication servers, the speaker is designed for applications including voice paging, loud ringing, and emergency/safety/security notification and alerting. Multiple SIP extensions can be configured to play a WAV file from memory or auto-answer for live voice paging. The 8186 supports both multicast receive and transmit for scalability and integration with IP phones and other Algo IP speakers, paging adapters and strobe lights. An integrated microphone supports talkback and ambient noise compensation for dynamic volume control. G.711 and G.722 wideband codecs are supported for optimum speech clarity and intelligibility. The 8186 is a self-amplified speaker and does not require any additional hardware or software.

As a SIP device, the 8186 eliminates the need for an ATA or FXS port. The 8186 appliance offers complete network management and supervision via a web interface, while also supporting common central provisioning protocols.

Integration in the UC environment means that the 8186 can be configured to interact with any device, phone or client that is connected to the communication network.

Multiple SIP extensions for RING and PAGE are available to register in the speaker, making the 8186 a multi-functional endpoint. On PAGE, the extension is designed to auto-answer for live voice page announcements. On RING, the extension is designed to play a WAV file stored in the device memory (e.g. loud ring tone, emergency alert announcement for lockdown, evacuation, medical, safety or security events, etc.). Several audio files are pre-loaded into the 8186 internal memory for ring sounds, however, custom WAV files can be easily uploaded to the speaker.

For multiple speaker deployments, multicasting (e.g. InformaCast, RTP, SA-Announce) permits solutions to be scaled over any size building, campus or enterprise in education, government, healthcare, commercial office, manufacturing, utility / plant, transportation, distribution / warehouse, and retail sectors, etc. Additional endpoints do not require a SIP license. A single registered speaker can be configured to multicast to any number and mix of additional Algo speakers, paging adapters, strobes lights and multicast capable IP phones as required, providing cost effective scalability using a minimum of network traffic. For multicast page applications, zone paging is easily supported to include any number of zones.

Wideband G.722 codec (HD voice) support in the speaker provides optimum speech clarity and intelligibility. Voice paging and emergency announcements can be easier to understand with wideband, particularly in a noisy environment. The wideband capability of the speaker improves STI-PA intelligibility scores, which is important in meeting NFPA 72 requirements and the NEMA SB-40 standard for emergency communication in education.

An integrated microphone in the speaker enables talkback. The microphone also has the capability to listen for ambient noise and automatically increase page volume when necessary. This feature enables a consistent ratio of page to ambient noise level without being unnecessarily loud.

The 8186 is CSA/UL, FCC and CE certified.

3 Executive Summary

The following summarizes tekVizion's findings:

- Test Case Failures:
  - None
- Test Cases Not Supported:
  - None
- Test Cases that are Not Applicable:
  - None
- Test Cases that were Not Executed:
  - None

- Observations:

  - Algo 8186 SIP Horn Speaker is manually registered with CUCM 12.0 using the Digest Authentication.
  - Algo 8186 SIP Horn Speaker has 2 types of Dial extensions.
    - Page extension: It provides announcements and talk back facility.
    - Ring extension: It is used for ring notification.
  - Algo 8186 SIP Horn Speaker has 3 working page modes
    - One Way mode: It has only one way audio path.
    - Two Way mode: It has 2 way audio path and enables talk back facility in the Algo 8186 horn speaker.
    - Delayed Page mode: It stores the page audio temporarily, and broadcasts it after the call is ended.

- G 722 codec is supported in Algo 8186 Horn Speaker.
- DUT can be a part of call functions like call Hold, Resume, Conference, Transfer etc. But these functions cannot be performed on the DUT.

4 Features Tested

- All test included in this test plan were executed unless otherwise noted.

4.2 Items Not Tested

Features that are specific to the internals of the 3rd party product or any features not listed will not be tested.

- All test included in this test plan were executed unless otherwise noted.

4.3 Assumptions

- Interoperability of 3rd party products – Testing will cover only features in 3rd party products that result in events to and/or from the CUCM or specified PSTN gateway.
- Call Processing – PSTN interface and Cisco SIP call processing traffic for all testing (excluding manual sampling run during traffic) may be generated using simulators.

5 Test Environment

5.2 Administration, Testing and Debugging tools

Tools used/required – Identify any tools required by 3rd party (partner under test). Also add Trace and Debug settings here.

Table 2 Administration, Testing and Debugging Tools

Product Name	Version	Type	Purpose	Units	Notes
Test Tools					
None					
3rd Party Tools					
None					
Debug Tools					
FileZilla Server	0.9.60	File server	For retrieving CDR from CUCM	1	Lab Provided

5.3 Equipment Requirements

Table below identifies all equipment/versions used for in this IVT.

Table 3 Equipment and Product Information

Product	Version	Type	Purpose	Units	Notes
Cisco Products					
CUCM	12.0	Call Server	Call Processing (For Local and Remote)	2	Lab Provided
CUPS	12.0	Presence Server	Presence and Messaging	1	Lab Provided
CIPC	8.6.1.0	Soft Phone	Endpoint	2	Lab Provided
Cisco Jabber	11.5.0	Soft Phone	Endpoint	1	Lab Provided
3rd Party Products					
Algo 8186 SIP Horn Speaker	1.6.1_rc1	Speaker	Endpoint	1	Customer provided

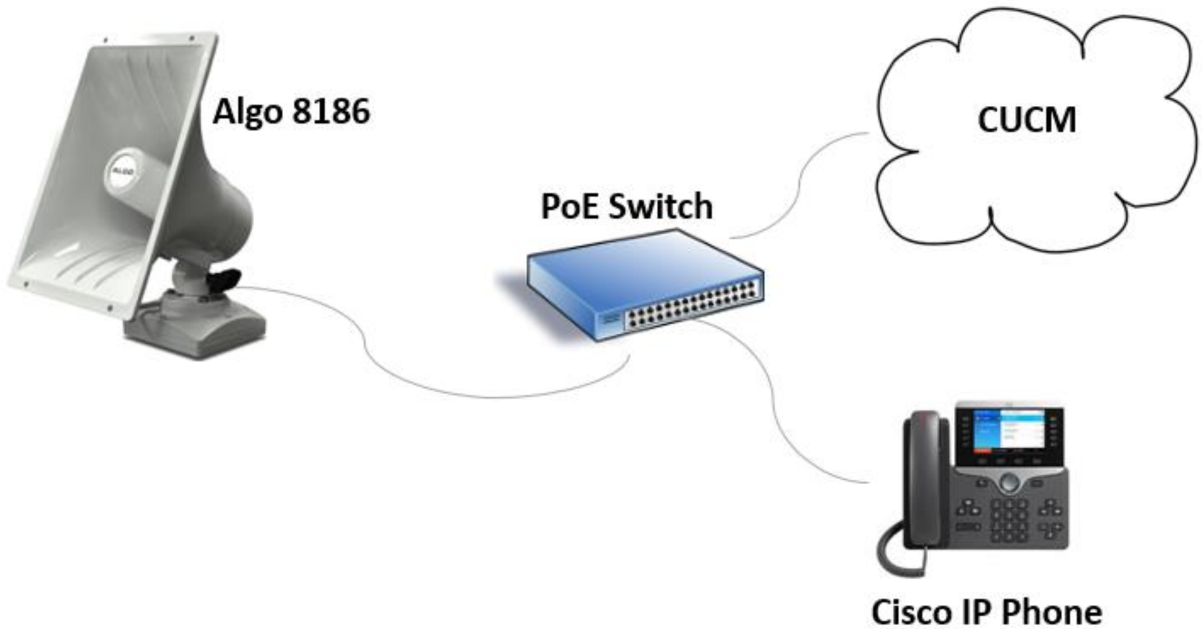
5.4 Cisco Phones

Table 4 Cisco Phones Information

Cisco Phone Model	Phone Firmware Version	Protocol	POE/Power	Units	Notes
7961	SIP41.9-4-2SR3-1S	SIP	POE	1	Lab Provided
7961	SCCP41.9-4-2SR3-1S	SCCP	POE	1	Lab Provided
8945	SIP894X.9-4-2SR3-1	SIP	POE	1	Lab Provided
7841	sip78xx.11-7-1-17	SIP	POE	2	Lab Provided
7975	SCCP75.9-4-2SR3-1S	SCCP	POE	1	Lab Provided

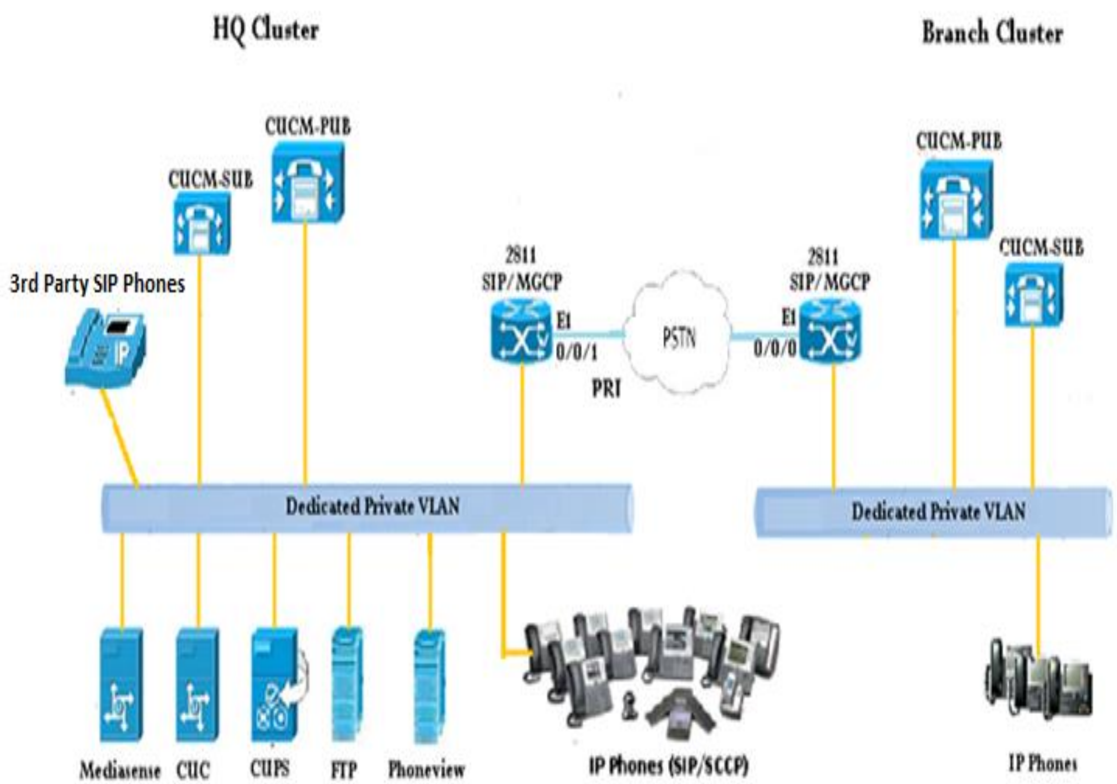
5.5 Deployment Architecture

Figure 1 – Deployment Architecture



5.6 Test Environment Architecture

Figure 2 – Test Environment Architecture



6 Test Cases

This section details the tests that will be performed during the testing period.

Result	Description
Pass	The test case passed with no exceptions
Fail	The test case failed – details of the failure are noted in the Comments column
N/A	The test case is not applicable to the product under test. Justification must be provided in the Comments column.
N/S	Not supported. While the feature tested by this test case generally would be considered a standard feature for this product category, this specific product (or this specific release) does not support the feature.
N/T	Not tested. The feature is supported by the product under test, but external factors (lab configuration, e.g.) prevented execution of the test. Justification must be provided in the Comments column.
Blocked	Other test case failures prevented the execution of this test. Reference to the corresponding failed test case must be provided in the Comments column.

6.1 Phase 1 Installation and Configuration Tests

Test is focused on ensuring that the 3rd party product (DUT) is registered with Call Manager successfully.

6.1.1 Register DUT to Cisco Call Manager

Test Case Details	
Title	Register DUT to Cisco Call Manager
Description	Verify 3rd party endpoints (DUT) are registered in Call Manager successfully
Test Setup	
Procedure	1. Connect the DUT in local CUCM cluster 2. Go to DUT(s) settings to verify network and load information 3. Associate end users to DUT(s) 4. Register DUT in local CUCM cluster with DN's
Expected Results	• CUCM Administration GUI display the DUT(s) • DUT is in "Registered" state • DUT(s) have a DN assigned • DUT(s) network data is correct: (VLAN, DNS, DHCP, TFTP, CUCM) • DUT(s) Phone Load version is correct • Users associated to DUT(s) respectively • DUT(s) registered to Local CUCM with assigned DN's
Observations	Pass DUT is registered in CUCM manually.

6.2 Phase 2 Functional Test

These tests test the various features of the 3rd party product and its various components. This involves the testing of the product against the Application note and IVT questionnaire requirements to ensure that it functions reliably and consistently in a manner that meets the requirements.

6.2.1 Intra-Cluster Calls

Test Case Details	
Title	Intra-Cluster Calls
Description	Verify intra-cluster calls between DUT, SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s): DUT(ring and page extension) • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1
Procedure	1. Local SIP Phone dials DUT ring extension and hear the ringing from the DUT 2. Local SIP Phone on-hook after 30s 3. Check for the audio. 4. Local SIP Phone dials DUT Page extension and DUT auto answers the call 5. Local SIP Phone on-hook after 30s 6. Check for the audio. 7. Local SCCP Phone dials DUT ring extension and DUT answers the call 8. Local SCCP Phone on-hook after 30s. 9. Check for the audio. 10. Local SCCP Phone dials DUT page extension and DUT answers the call 11. Local SCCP Phone on-hook after 30s. 12. Check for the audio. 13. Retrieve CDR from CUCM 14. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 4 calls establish with 2-way audio path

	• Calling and Called Parties hear ring-back and ring tone • 4 calls terminate normally
Observations	Pass DUT auto answer the call.

6.2.2 Inter-Cluster Call

Test Case Details	
Title	Inter-Cluster Call
Description	Verify inter-cluster calls between DUT(s), SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT Remote CUCM ➢ SCCP: Remote SCCP Phone1 ➢ SIP: Remote SIP Phone1
Procedure	1. Remote SIP Phone dials DUT and DUT answers the call 2. Remote SIP Phone on-hook after 30s. 3. Check for the audio. 4. Remote SCCP Phone dials DUT and DUT answers the call 5. Remote SCCP Phone on-hook after 30s. 6. Check for the audio. 7. Calling & Called party release calls alternatively 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 calls establish with audio path • Calling and Called Parties hear ring-back and ring tone • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass DUT auto answer the call.

6.2.3 One Way Page mode

Test Case Details	
Title	One Way Page mode
Description	Verify One Way Page mode calls between DUT(s), SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1
Procedure	• Local SIP Phone dials DUT • DUT answers the call • Local SIP Phone on-hook after 30s. • Check for the audio. • Local SCCP Phone dials DUT and DUT answers the call • Local SCCP Phone on-hook after 30s. • Check for the audio. • Calling & Called party release calls alternatively. • Retrieve CDR from CUCM • Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 calls established with one way audio path • Calling and Called Parties hear ring-back and ring tone • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass DUT auto answer the call.

6.2.4 Two Way Page mode

Test Case Details	
Title	Two Way Page mode
Description	Verify Two Way Page mode calls between DUT(s), SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1
Procedure	1. Local SIP Phone dials DUT and DUT answers the call 2. Local SIP Phone on-hook after 30s. 3. Check for the 2-way audio. 4. Local SCCP Phone dials DUT and DUT answers the call 5. Local SCCP Phone on-hook after 30s. 6. Check for the 2-way audio. 7. Calling & Called party release calls alternatively 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
	• 2 calls establish with bi-directional audio path • Calling and Called Parties hear ring-back and ring tone • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.5 Delayed Page mode

Test Case Details	
Title	Delayed Page mode
Description	Verify Delayed Page mode calls between DUT(s), SCCP and SIP endpoints
Test Setup	Local CUCM • DUT(s):DUT • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1
Procedure	1. Local SIP Phone dials DUT and DUT answers the call 2. Check for the audio. 3. Local SIP Phone on-hook after 30s. 4. Local SCCP Phone dials DUT and DUT answers the call 5. Local SCCP Phone on-hook after 30s. 6. Check for audio. 7. Calling & Called party release calls alternatively 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
	• Call established between Local SIP phone1 and DUT and after call termination the DUT plays the audio. • Call established between Local SCCP phone1 and DUT and after call termination the DUT plays the audio. • 2 calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass After hearing ring back tone from DUT user should give an announcement or voice message, when the cisco IP phone goes on hook the DUT will give the announcement. The DUT stores the voice message for some time temporarily. Maximum Page timeout is 5 minutes.

6.2.6 Off-Net Calls

Test Case Details	
Title	Off-Net Calls
Description	Verify basic calls between DUT(s) and PSTN endpoints
Test Setup	• Local CUCM ➔ DUT(s):DUT • PSTN Phone➔ PSTN (SIP);
Procedure	1. PSTN dials DUT and DUT answers the call 2. PSTN on-hook after 60s 3. Retrieve CDR from CUCM Server 4. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call established with audio path • Calling and Called Parties hear ring-back and ring tone • 2 Calls terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.7 Functional Test: CFA

Test Case Details	
Title	Functional Test: CFA
Description	Verify "CFA" calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	Local CUCM • DUT(s):DUT • SCCP: Local SCCP Phone1 • SIP: Local SIP Phone1 Remote CUCM • SCCP: Remote SCCP Phone1 • SIP: Remote SIP Phone1 PSTN Phone: PSTN Enable CFA for DN(s): · Device➔Phone➔DUT➔CFA➔Local SCCP Phone1 (SCCP) · Device➔Phone➔ DUT ➔CFA➔ Local SIP Phone1

	· Device➔Phone➔Local SCCP Phone➔CFA➔DUT(DUT) · Device➔Phone➔PSTN➔DUT (DUT)
Procedure	1. Remote SIP phone1 dials DUT, Local SCCP phone1 answers and Local SCCP phone1 on-hook after 30s 2. Remote SCCP phone 1 dials DUT, Local SIP phone answers and , Local SIP Phone1 answers and Local SIP Phone1 on-hook after 30s 3. Remote SIP phone1 dials Local SCCP phone1, DUT answers and Remote SIP phone 1 on-hook after 30s 4. Retrieve CDR from CUCM 5. Check Calling, Called, Duration, Origination & Termination Cause Codes **Note:** Upon test completion, remove “CFA” feature for devices before proceeding to next test case CUCM Administration GUI: Device➔Phone➔DN➔Line➔Call Forward All ➔Destination➔blank
Expected Results	• Call forward to Local SCCP phone1 and phone rings • Call establish between Local SIP phone1 & Local Sip phone with 2-way audio • Call terminate normally • Call forward to Local SIP phone1 and phone rings • Call establish between Local SCCP phone1 & Local SIP phone with 2-way audio • Call terminate normally • Call forward to DUT and phone rings • Call establish between Local SIP phone1 & DUT with 2-way audio • Call terminate normally • 6 CDR(s) retrieved • Selected CDR(s) fields match calls
Observations	Pass

6.2.8 Functional Test: CFNA

Test Case Details	
Title	Functional Test: CFNA
Description	Verify “CFNA” calls between DUT(s), SCCP and SIP endpoints

Test Setup	• Local CUCM
	  · DUT(s):DUT
	  · SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
	• Remote CUCM
	  · SCCP: Remote SCCP Phone1; SIP: Remote SIP Phone1;
	• Voicemail and Call Waiting disabled for all DN(s) Device➔Phone➔DN➔Line➔
	  · Voicemail➔No Voicemail
	  · Call Waiting➔Max. Calls➔1; Busy Trigger➔1;
	
	Enable CFNA for DN(s):
	· Device➔Phone➔DUT➔CFNA➔Local SCCP Phone1 (SCCP)
	· Device➔Phone➔DUT➔CFNA➔Remote SIP Phone1 (SIP)
	· Device➔Phone➔Local SCCP Phone1➔CFNA➔DUT (DUT)

Procedure	1. Remote SIP phone1 dials DUT, DUT does not answer and Local SCCP phone1 answers 2. Local SCCP phone1 on-hook after 30s 3. Remote SIP phone1 dials DUT, DUT does not answer and Local SIP phone1 answers 4. Local SIP phone1 on-hook after 30s 5. Remote SIP phone1 dials Local SIP phone 1 , Local SIP phone 1 does not answer and DUT answers 6. Remote SIP phone1 on-hook after 30s 7. Retrieve CDR from CUCM 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call forward to Local SCCP phone1 after ring timeout • Call establish between Remote SIP phone 1 & Local SCCP phone1 with 2-way audio • Call terminate normally • Call forward to Local SIP phone1 after ring timeout • Call establish between Remote SCCP phone 1 & Local SIP phone1 with 2-way audio

	• Call terminate normally • Call forward to DUT after ring timeout • Call establish between Remote SIP phone 1 & DUT with 2-way audio • Call terminate normally • CDR(s) retrieved • Selected CDR(s) fields match calls
Observations	Pass

6.2.9 Functional Test: CFB

Test Case Details	
Title	Functional Test: CFB
Description	Verify "CFB" calls between DUT(s), SCCP and SIP endpoints
Test Setup	• Local CUCM · DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM SCCP: Remote SCCP Phone1; SIP: Remote SIP Phone1; • Voicemail and Call Waiting disabled for all DNs Enable CFB for DN(s): · Device➔Phone➔DUT➔CFB➔Local SCCP Phone1 (SCCP) · Device➔Phone➔Local SCCP Phone1➔CFB➔DUT
Procedure	1. Remote SIP phone1 dials DUT and DUT answers the call 2. Local SIP phone 1 dials the DUT, Local SCCP phone 1 answer the call. 3. Remote SIP phone 1 goes on hook. 4. Local SIP phone 1 goes on hook. 5. Remote SCCP phone dials local SCCP phone 1 and Local SCCP phone 1 answers 6. Local SIP phone 1 dials local SCCP phone 1, DUT answer the call. 7. Remote SCCP phone 1 goes on hook. 8. Local SIP phone 1 goes on hook. 9. Retrieve CDR from CUCM

	10. Check Calling, Called, Duration, origination & termination cause codes matches the calls
Expected Results	• Call establish between Remote SIP Phone1 & DUT with 2-way audio • Call forward to Local SCCP phone 1 and phone rings • Call establish between Local SIP phone 1 & Local SCCP phone 1 with 2-way audio • Remote SIP phone 1end the call. • Local SCCP phone 1 end the call. • Call establish between Remote SCCP Phone1 & local SCCP phone 1 with 2-way audio • Call forward to DUT and DUT answer the call. • Call establish between Local SIP phone 1 & DUT with 2-way audio • Remote SIP phone 1end the call. • Local SCCP phone 1 end the call. • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.10 Functional Test: Hold & Resume

Test Case Details	
Title	Functional Test: Hold & Resume
Description	Verify “Hold & Resume” calls between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • PSTN: PSTN • Remove all CFA, CFNA & CFB settings on DN(s) used in previous test cases
Procedure	1. Local SCCP Phone1 dials DUT, DUT answers and Local SCCP Phone1 hits “Hold” after 20s 2. Local SCCP Phone1 hits “Resume” after 20s and Local SCCP Phone1 on-hook after 30s

	3. Local SIP Phone1 dials DUT, DUT answers incoming call and Local SIP Phone1 on-hold 4. Local SIP Phone1 hits “Resume” after 60s and DUT on-hook after 30s 5. Repeat steps 1-2 for PSTN. Replace Local SCCP phone 1 with PSTN 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SCCP Phone1 & DUT with 2-way audio • DUT is On-Hold (MOH) • Call resume between Local SCCP Phone1 & DUT • Call terminate normally • Call establish between Local SIP Phone1 & DUT with 2-way audio • DUT is On-Hold (MOH) • Call resume between Local SIP phone 1 & DUT • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.11 Functional Test: Blind Transfer

Test Case Details	
Title	Functional Test: Blind Transfer
Description	• Verify “Blind Transfer” calls between DUT(s), SIP and SCCP endpoints
Test Setup	• Local CUCM→DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • PSTN: PSTN
Procedure	1. Local SIP phone 1 dials Local SCCP phone 1, Local SCCP phone 1 answers and hits “Transfer” after 30s

	2. Local SCCP phone 1 dials DUT, Local SCCP phone1 hits "Transfer" and Local SCCP phone1 is on-hook 3. DUT answer the call 4. Local SIP phone 1 goes on-hook after 60s 5. PSTN dials Local SCCP phone 1, Local SCCP phone 1 answers and hits "Transfer" after 30s 6. Local SCCP phone 1 dials DUT and hits "Transfer" and Local SCCP phone1 is on-hook 7. DUT answer the call. 8. PSTN goes on-hook after 60s 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & Local SCCP phone 1 with 2-way audio • Local SIP phone 1 is On-Hold (MOH) • Local SCCP phone 1 blind transfer to DUT with 2-way audio path • All calls terminate normally • Call establish between PSTN & Local SCCP phone 1 with 2-way audio • PSTN is On-Hold (MOH) • Local SCCP phone 1 blind transfer to DUT • All calls terminate normally • 4 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.12 Functional Test: Consult Transfer

Test Case Details	
Title	Functional Test: Consult Transfer
Description	Verify "Consult Transfer" calls between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM→DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • PSTN: PSTN
Procedure	1. Local SIP phone 1 dials Local SCCP phone 1, Local SCCP phone 1 answers and hits "Transfer" after 30s 2. Local SCCP phone 1 dials DUT, And DUT answers the call 3. Local SCCP phone1 hits "Transfer" and Local SCCP phone1 is on-hook 4. Local SIP phone 1 goes on-hook after 60s 5. PSTN dials Local SCCP phone 1, Local SCCP phone 1 answers and hits "Transfer" after 30s 6. Local SCCP phone 1 dials DUT, and DUT answer the call. 7. Local SCCP phone1 hits "Transfer" and Local SCCP phone1 is on-hook 8. PSTN goes on-hook after 60s 9. Check the Calling, Called, Duration, Origination & Termination Cause Codes 10. Retrieve CDR from CUCM 11. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & Local SCCP phone 1 with 2-way audio • Local SIP phone 1 is On-Hold (MOH) • Call establish between Local SCCP phone 1 & DUT with 2-way audio • Local SCCP phone 1 transfer to DUT with 2-way audio path • All calls terminate normally • Call establish between PSTN & Local SCCP phone 1 with 2-way audio • PSTN is On-Hold (MOH)

	• Call establish between Local SCCP phone 1 & DUT with 2-way audio • Local SCCP phone 1 transfer to DUT • Local SCCP phone 1 hears reorder tone • All calls terminate normally • 6 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.13 Functional Test: Conference Call

Test Case Details	
Title	Functional Test: Conference Call
Description	Verify Conference call between DUT(s), SIP, SCCP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Service parameter: Drop Ad Hoc Conference ➔ Never (Default) • Media Resource Group (MRG) & Media Resource Group List (MRG_L) • Assign Media Resource: System➔Device Pool➔ep_pool➔Media Resource Group List➔MRG_L
Procedure	1. Local SCCP Phone1 dials Local SIP Phone1, Local SIP Phone1 answers and DUT hits “Conference” after 30s 2. Local SIP Phone1dials DUT, DUT answers 3. Local SIP Phone1 hits “Conference” after 30s 4. Local SCCP Phone1 goes on-hook after 60s 5. Local SIP Phone1goes on-hook after 30s 6. PSTN dials Local SIP Phone1, Local SIP Phone1 answers and hits “Conference” after 30s 7. Local SIP Phone1dials DUT, DUT answers 8. Local SIP Phone1 hits “Conference” after 30s 9. PSTN goes on-hook after 60s 10. Local SIP Phone1goes on-hook after 30s 11. Retrieve CDR from CUCM 12. Check the Calling, Called, Duration, Origination & Termination Cause Codes

Expected Results	• Call establish between Local SCCP Phone1 & Local SIP Phone1 with 2-way audio • Local SCCP Phone1 is On-hold (MOH) • Local SIP Phone1 is conference-in • 3 parties in conference call with 3-way audio • Local SCCP Phone1 left conference. Local SIP Phone1 & DUT connect directly • All calls terminate normally • Call establish between PSTN & Local SIP Phone1 with 2-way audio • PSTN is On-Hold (MOH) • PSTN, Local SIP Phone1 & DUT is conference-in • All 3 parties in conference call with 3-way audio • PSTN leave conference. Local SIP Phone1 & DUT connect directly • All calls terminate normally • CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.14 Functional Test: Jabber for Windows

Test Case Details	
Title	Functional Test: Jabber for Windows
Description	Verify Jabber calls originating & terminating to DUT(s) endpoints (Jabber for Windows)
Test Setup	• Local CUCM➔DUT(s):DUT SCCP: Local SCCP Phone1; • Jabber for Windows (Device➔Phone➔Add New➔CSFUSER1:DN:1922; End User:juser01/123456) • Windows PC with Jabber clients installed
Procedure	1. 1922 dials DUT (Duration=30s) 2. 1922 dials Local SCCP phone 1 (Duration=30s) 3. Local SCCP phone 1 dials DUT (Duration=30s) 4. Calling and Called party goes on-hook alternatively 5. Retrieve CDR from CUCM

	6. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 3 calls establish with 2-way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.15 Functional Test: IP Communicator

Test Case Details	
Title	Functional Test: IP Communicator
Description	Verify IP Communicator calls originating & terminating to DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT, SIP: Local SIP Phone1; CIPC:1940; • Launch IP Communicator on a Windows PC
Procedure	1. 1940 dials DUT (Duration=30s) 2. 1940 dials Local SIP phone 1 (Duration=30s) 3. Local SIP phone 1 dials DUT(Duration=30s) 4. Calling and Called party goes on-hook alternatively 5. Retrieve CDR from CUCM 6. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 3 calls establish with 2-way audio • 3 calls terminate normally • 3 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.2.16 Functional Test: Mobile Voice Access (MVA)

Test Case Details	
Title	Functional Test: Mobile Voice Access (MVA)
Description	Verify Inbound Mobile Voice Access (MVA) calls from DUT(s) endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Remote CUCM ➔DUT:DUT; • CUCM Service Parameter: · Enable Enterprise Feature Access➔True · Enable Mobile Voice Access➔True · Mobile Voice Access Number➔8005555 · Matching Caller ID with Remote Destination➔ Partial Match · Number of Digits for Caller ID Partial Match➔7 • Mobile Voice Access service running on CUCM-PUB • Mobile Voice Access enabled on Voice Gateway • MVA # provisioned in CUCM: Media Resources➔Mobile Voice Access➔Add New➔8005555 • Local Cluster Single Number Reach (SNR) configured for Local SIP phone 1 ➔Remote Device:234- Local SIP phone 2: · Add Remote Destination Profile:Device➔Device Settings➔Remote Destination Profile➔Add New➔mobile1_rdp ➢ Userid:dutuser02 ➢ Add New Remote Destination: · Name➔Mobility_1 · Destination Number➔ 234 Local SIP phone 2 · Check “Enable Unified Mobility”, “Enable Single Number Reach” & “Enable Move to Mobile” · Device Settings➔Softkey Template➔SIP_EP_User➔Add “Mobility” softkey (On-Hook & Connected) · User Management➔End User➔dutuser02➔Check ”Enable Mobility” &”Enable Mobile Voice Access” · Device➔Phone➔Local SIP phone➔Owner userid➔user02

	· Remote Device: Device➔Phone➔ Local SIP phone 2 ➔Line➔No Answer Ring Duration➔60
Procedure	1. (Mobile device) dial MVA #8005555 2. Mobile User enters PIN:123456# & DN:DUT# 3. Mobile User dial DUT# 4. DUT answers 5. Mobile device sends DTMF *74 to handoff session after 30s 6. DUT goes on-hook after 60s 7. Retrieve CDR from CUCM 8. Check the Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Mobile user prompted for PIN & destination DUT # • DUT is ringing • Call establish between Mobile device & DUT with 2-way audio • Call hand-off to DUT • Call terminate normally • 1 CDR retrieved • Selected fields in CDR matches call
Observations	Pass

6.3 Phase 3 Negative Tests

These tests are executed to determine the ability of the impact on calls, the CUCM and the 3rd party application when combinations of the aforementioned fail by power failure or network connectivity problems. Testing robustness of the application through hardware and software fault insertion i.e. Failover/fallback.

6.3.1 Negative Test: PUB Failure

Test Case Details	
Title	Negative Test: PUB Failure
Description	Verify a PUB failure should not affect stable or transient calls on DUT(s)
Test Setup	• Local CUCM➔DUT(s):DUT ;SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Procedure	1. Local SIP phone 1 dials DUT, DUT answers

	2. Local SCCP Phone1 dials Local SIP phone 2, Local SIP phone 2 answers 3. Access CUCM-PUB server via SSH (Local Cluster) 4. Enter CLI: utils system restart <CR> yes 5. Call terminates normally. 6. Local SCCP Phone1 dials DUT, DUT answers 2nd incoming call 7. Called party goes on-hook for all 3 calls 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & DUT with 2-way audio • Call establish between Local SCCP Phone1 & Local SIP phone 2 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between Local SCCP Phone1 & DUT with 2-way audio • Transient calls not impacted by PUB restart • All calls terminate normally • CUCM-PUB is in-service • All calls successful after PUB failure recovery • 3 CDR(s) retrieved • Selected fields in CDR matches calls
Observations	Pass

6.3.2 Negative Test: SUB Failure

Test Case Details	
Title	Negative Test: SUB Failure
Description	Verify a SUB failure should not affect stable calls on DUT(s)
Test Setup	• Local CUCM➔DUT(s):DUT;SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Procedure	1. Local SIP phone 1 dials DUT, DUT answers 2. Local SCCP Phone1 dials Local SIP phone 2, Local SIP phone 2 answers

	3. Access CUCM-SUB server via SSH (Local Cluster) 4. Enter CLI: utils system restart <CR> yes 5. Call terminates normally. 6. Local SCCP Phone1 dials DUT, DUT answers 2nd incoming call 7. Called party goes on-hook for all 3 calls 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & DUT with 2-way audio • Call establish between Local SCCP Phone1 & Local SIP phone 2 with 2-way audio • CUCM-PUB is restarted • Stable calls not impacted by PUB restart • Call establish between Local SCCP Phone1 & DUT with 2-way audio • Transient calls not impacted by PUB restart • All calls terminate normally • CUCM-PUB is in-service • All calls successful after PUB failure recovery • 3 CDR(s) retrieved • Selected fields in CDR matches calls
Observations	Pass Call cannot be hanged up in DUT during the preservation mode.

6.3.3 Negative Test: Phone Network Failure

Test Case Details	
Title	Negative Test: Phone Network Failure
Description	Verify DUT(s) recovers from a network failure
Test Setup	• Local CUCM➔DUT(s):DUT;SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Procedure	1. Local SIP phone 1 dials DUT, DUT answers 2. Unplug network cable from device DN:DUT 3. Restore the network cable after 60s 4. Local SIP Phone1 dials DUT, DUT answers 5. DUT goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & DUT with 2-way audio • Network failure reported on device DN:DUT • Stable call drops • Device DUT re-registers after network cable restored • Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between Local SIP Phone1 & DUT with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.3.4 Negative Test: Phone Power Failure

Test Case Details	
Title	Negative Test: Phone Power Failure
Description	Verify DUT(s) recovers from a power failure
Test Setup	• Local CUCM➔DUT(s):DUT;SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Procedure	1. Local SIP phone 1 dials DUT, DUT answers 2. Remove power cable from DUT 3. Restore power cable after 60s 4. Local SIP Phone1 dials DUT, DUT answers call 5. Local SIP phone 1 goes on-hook after 60s 6. Retrieve CDR from CUCM 7. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & DUT with 2-way audio • DUT lost power • Stable call drops • Device DUT re-registers after power is restored • Network Data: DNS, DHCP, TFTP, CUCM, VLAN, Load ID are restored on device • Call establish between Local SIP Phone1 & DUT with 2-way audio • Call terminate normally • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.4 Phase 4 Miscellaneous Tests

These tests are executed to verify specific information about the third-party products provided by partners

6.4.1 Miscellaneous Test: Codec (G722)

Test Case Details	
Title	Miscellaneous Test: Codec (G722)
Description	Verify URI calls between DUT(s) & SIP endpoints for In-band Codec (G722)
Test Setup	• Local CUCM➔DUT(s): DUT. SCCP: Local SCCP Phone1 ; SIP: Local SIP Phone1 ; • Go to System➔Region Information➔ Audio Codec Preference List➔ Add New➔ G722➔Select G722 Codec • Go to System➔Region Information➔ Region➔ Add New➔G722-Region➔G722 • Go to System➔Device Pool➔ Add New➔G722-dp➔Region➔G722-Region • Update DUT with device pool=G722
Procedure	1. Local SIP phone 1 hits dials DUT 2. DUT answers call 3. Local SIP phone 1goes on-hook after 60s 4. Local SCCP phone 1 hits dials DUT 5. DUT answers call 6. Local SCCP phone 1goes on-hook after 60s 7. Retrieve CDR from CUCM Server 8. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• 2 calls establish with 2 way audio for G722 codec • 2 calls terminate normally • Voice quality was good for codec type • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass

6.4.2 Long Duration Calls

Test Case Details	
Title	Long Duration Calls
Description	Verify long duration calls between DUT(s), SCCP, SIP and PSTN endpoints
Test Setup	• Local CUCM➔DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1;
Procedure	1. Local SIP phone 1 dials DUT, DUT answers (Duration: 1 Hr.) 2. Local SCCP Phone1 dials DUT, DUT answers (Duration: 1 Hr) 3. Retrieve CDR from CUCM 4. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Call establish between Local SIP phone 1 & DUT with 2-way audio • Call establish between Local SCCP Phone1 & DUT with 2-way audio • All long duration calls were stable with 2-way audio • 2 CDR(s) retrieved • Selected fields in CDR(s) match calls
Observations	Pass Page timeout should be select for 60 mins.

6.4.3 Miscellaneous Test: Cisco Phone Models

Test Case Details	
Title	Miscellaneous Test: Cisco Phone Models
Description	Verify calls and mid-call features between DUT(s) and various Cisco IP Phone Models
Test Setup	• Local CUCM→DUT(s):DUT; SCCP: Local SCCP Phone1; SIP: Local SIP Phone1; • Cisco Phone Models:8861, 8945, 7975, 9971,7842
Procedure	1. Cisco IP Phone dials DUT, DUT answers and Cisco IP Phone on-hook after 120s 2. Cisco IP Phone dials DUT , DUT answers and Cisco IP Phone hits “Hold” after 20s 3. Cisco IP Phone hits “Resume” after 20s, Cisco IP Phone on-hook after 120s 4. Cisco IP Phone dials Local SIP phone 1, Local SIP phone answers and Cisco IP Phone hits “Transfer” after 20s 5. Cisco IP Phone dials DUT, Cisco IP Phone hits “Transfer and Cisco IP Phone on-hook 6. Local SIP phone 1 goes on-hook after 120s 7. Repeat steps 1-6 by replacing DN: Cisco IP Phone with DN(s) of other Cisco phone models 8. Retrieve CDR from CUCM 9. Check Calling, Called, Duration, Origination & Termination Cause Codes
Expected Results	• Calls establish between DUT & Cisco IP Phone • Call Hold/Resume between DUT & Cisco IP Phone • Blind Transfer between DUT & Cisco IP Phone • Conference Call between DUT & Cisco IP Phone • CDR(s) retrieved for all the calls • Selected fields in CDR(s) match calls
Observations	Pass